Abstract—The paper compares the performance of the Block Linear Equalizer used with the Block Data Transmission System with that of the conventional Linear Transversal Equalizer that is used with an Uninterrupted Continuous System. The result of the comparison show that the Block Data Transmission System has some useful advantages over the Continuous System like exact channel equalization is achieved in all cases and the system is free from error extension effects. The disadvantage of the block Data Transmission System is that for a given information rate the bandwidth required is larger as compared to the continuous systems. This reduces the tolerance to noise of the former systems and partially offsets the advantages gained by the system. However, it is seen that when the number of signals in a group is large as compared to the number of symbols set to zero, the performance of the Block Linear Equalizer is similar to that of the Transversal Equalizer.

Index Term—Linear Transversal Equalizer, Block Detection, Wireless Channel, Inter-symbol Interference.

I. INTRODUCTION

In the study of distorted digital signals techniques of both linear and nonlinear equalization of the channel have been widely studied [1,2,3]. The non-linear equalization of the channel usually gives a better tolerance to additive white Gaussian noise than linear equalization, normally requiring a lower average signal to noise power ratio for a given error rate. An even better tolerance to noise can be achieved through the use of more sophisticated detection processes which do not equalize the channel. Many of these processes, however, involve considerable equipment complexity [4,5].

An interesting technique called the Block Data Transmission system (BDTS) has recently been proposed which for certain applications can achieve a similar standard of performance as the more sophisticated processes just mentioned, but with relatively simple equipment [5-8]. The system is a synchronous serial data transmission system and employs transmission of alternating blocks of data and training/zero valued symbols. In contrast to the recursive symbol-by-symbol detection approach usually employed, each data block is here detected as a unit. The Block Linear Equalizers (BLE) and Block Decision Feed Back Equalizers (BDFE) have been designed for the system and the system has been used in HF Modems [6,7,8].

For a given information rate, the element transmission rate in the arrangement of the BDTS just considered is increased as compared to the systems where the signal-elements are transmitted in a continuous (uninterrupted) stream. For a given transmission path, this normally results in less intersymbol interference in the sample values of the received signal in the case of the Continuous Systems. This paper compares the noise performance of the conventional Linear Transversal Equalizer (LTE) that is used with Continuous System with that of the Block Linear Equalizer (BLE) that is used with the Block Data Transmission System. The results of the comparison show that despite of the increased level of intersymbol interference in the BLE it provides a similar standard of performance as the LTE. Furthermore, the BDTS possesses some basic advantages and provides a better alternative to Continuous systems without any significant increase in equipment complexity.

II. MODEL OF THE DATA TRANSMISSION SYSTEM

Fig. 1 shows the model of the data transmission system considered. Each impulse $s_i \delta(t - iT)$ at the input to the channel is the corresponding input signal-element and it may be either binary or multilevel. The signal elements are assumed to be antipodal and statistically independent.

![Fig. 1. Model of the Data Transmission System](image)

The linear baseband channel has an impulse response $h(t)$ and includes all transmitter and receiver filters used for pulse shaping and linear modulation and demodulation. The impulse response $h(t)$ of the transmitter and receiver filters in cascade is assumed to be such that $h(0) = 1$ and $h(iT) = 0$ for all nonzero integer values of $i$. This impulse response is achieved by using the same transfer function $B(f) = H(f)^{1/2}$ for the transmitter and receiver filters, where

$$H(f) = \begin{cases} \frac{1}{2} T \left[ 1 + \cos \frac{T}{2} f \right] & \text{for } -\frac{1}{T} < f < \frac{1}{T} \\ 0 & \text{elsewhere} \end{cases}$$

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White Gaussian noise is introduced at the output of the transmission path. The noise has zero mean and a two sided power spectral density of $\sigma^2$, giving the zero mean Gaussian waveform $w(t)$ at the output of the receiver filter. Thus the resultant signal at the output of the receiver filter is

$$r(t) = \sum \limits i Si y(t-iT) + w(t)$$  \hspace{1cm} (2)

The received signal at the output of the receiver filter is sampled at time instant $t = iT$, for all integers $i$. $T$ being the symbol interval.

The signal distortion introduced by the transmission path is assumed to be such that a received signal-element may introduce inter-symbol interference in the sample values of some or all of the $g$ immediately following elements. The sampled impulse response of the base band channel (i.e. the transmitter filter, transmission path and the receiver filter in cascade) is

$$\sum \limits h=0 \limits ^{h} Y(h) \delta (t - hT)$$  \hspace{1cm} (3)

where $y = y(hT)$ is now non-zero for some or all values of the integer $h$ in the range 0 to $g$ and is zero for all other values of $h$. The sampled impulse response of the channel may simply be written as the $(g+1)$-component row vector

$$y_0 \ y_1 \ y_2 \ \cdots \ y_g$$  \hspace{1cm} (4)

When a continuous stream of signal-elements is received in the presence of noise then neglecting the delay introduced by the filters for convenience, the sample value of the received signal at time $t = iT$ is

$$r_h = \sum \limits i=h-g \limits ^{h} S_i Y_{h-i} + W_h$$  \hspace{1cm} (5)

where it can be shown that $w$ are sample value statistically independent Gaussian random variables of zero mean and variance $\sigma^2$ [1,2,3].

With the transmission of a continuous (uninterrupted) stream of signal-elements, the $i$th sample value of the received signal is

$$r_i = \sum \limits j=0 \limits ^{g} Y_{i-j} S_i + W_i$$  \hspace{1cm} (6)

In the presence of delay and distortion in transmission, if $s_i$ is detected from $r_i$, there is, in addition to the noise component $w_i$, an inter-symbol interference component

$$\sum \limits j=1 \limits ^{g} Y_{j} S_{i-j}$$  \hspace{1cm} (7)

added to the wanted signal $y_0$, $s_i$. The data transmission system in Figure 1 must now include an equalizer at the receiver, which equalizes the baseband channel.

III. THE LINEAR TRANSVERSAL EQUALIZER

A linear equalizer is usually a feed forward transversal filter [1-4]. Assume that the filter has $p$ taps and the $i$th tap has a gain $d_{i,1}$ so that the tap gains of the filter can be represented by the $p$-component row-vector

$$D = [d_0 \ d_1 \ \cdots \ d_{p-1}]$$  \hspace{1cm} (8)

The sampled impulse response of the equalized channel is obtained by the convolution of the sampled impulse response of the baseband channel and that of the filter. It is represented by the $(p+g)$-component row-vector

$$E = [e_0 \ e_1 \ \cdots \ e_{p+g-1}]$$  \hspace{1cm} (9)

Let the desired or the ideal value of the equalized channel be represented by the $(p+g)$-component vector $U_h$ for a total transmission delay of $hT$ seconds. Where

$$U_h = \begin{bmatrix} 0 \ldots 0 & 1 & 0 \ldots & 0 \end{bmatrix}$$  \hspace{1cm} (10)

It has been shown [1-3] that the linear equalizer that achieves the most effective equalization of the channel for a given number of tap gains is the one that minimizes the expression

$$|U_h - E|^2$$  \hspace{1cm} (11)

i.e. the mean square error between the desired and the actual value of the equalized channel. Furthermore, the sampled impulse response of the equalizer is given by the components of the $p$-component row vector

$$D = U_h (BB^T)^{-1}$$  \hspace{1cm} (12)

$B$ is a $p \times (p+g)$ matrix whose $i$th row is

$$B_i = \begin{bmatrix} 0 \ldots 0 & y_{i,0} \ldots y_{i,g} & 0 \ldots 0 \end{bmatrix}$$  \hspace{1cm} (13)

$D$ must now be determined for each value of $h$ in the range 0 to $p+g-1$ and the vector $D$ for the required equalizer is that which gives the minimum value of $|U_h - E|$. Thus when $|U_h - E| \ll 1$ the output of the linear equalizer at the time instant $(i+h)T$ is

$$x_{i+h} = s_i + v_{i+h}$$

$v_{i+h}$ is a sample value of a Gaussian random variable with zero mean and variance

$$\eta^2 = \sigma^2 \sum \limits i=0 \limits ^{p-1} d_i^2 = \sigma^2DD^T$$  \hspace{1cm} (14)

and $s_i$ is detected by comparing $x_{i+h}$ with the appropriate thresholds. Methods of implementing the linear equalizer for both time invariant and time varying channels have been

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IV. THE BLOCK LINEAR EQUALIZER

In the block transmission system considered here consecutive blocks of ‘m’ information symbols at the input to the transmitter filter are separated by blocks of ‘g’ zero-level symbols. The data-transmission system of Figure 1, considered earlier, is now modified as follows. Following a group of m impulses, at the input to the transmitter filter, the next g impulses are set to zero, so that adjacent groups of m transmitted signal-elements are separated by g zero-level elements [5-8]. The zero-level elements form gaps (time guard bands) between adjacent groups of transmitted signal-elements and so prevent inter-symbol interference between the corresponding groups of elements at the receiver input.

Associated with each received group of m signal elements there are

\[ n = m + g \] (15)

sample values of the received signal, which depend on these elements and is independent of the other received elements. A received group of m signal elements is detected from the corresponding n sample values. The n sample values used for the detection of a received group of m signal elements are given by the n components of the row vector

\[ R = \sum_{i=1}^{m} S_i Y_i + W = SY + W \] (16)

where \( S = \{s_i\} \) and \( W = \{w_i\} \) are m- and n-component row vectors, respectively, and the \( \{w_i\} \) are sample values of statistically independent Gaussian random variables with zero mean and variance \( \sigma^2 \). \( Y \) is an m x n matrix of rank m whose ith row \( Y_i \) is

\[ Y_i = \begin{bmatrix} y_{1i} & y_{2i} & \cdots & y_{ni} \end{bmatrix} \] (17)

It has been shown that when the detector has no prior knowledge of S or \( \sigma^2 \), the best linear estimate it can make of S from the received vector R is the m-component row vector \( X \), such that \( XY \) is the point in the m-dimensional subspace spanned by the m \( \{Y_i\} \) at the minimum distance from R [8]. By the projection theorem, \( XY \) is the orthogonal projection of R on to the m-dimensional subspace, so that \( [R-XY]Y^T = 0 \) or

\[ X = R Y^T (yy^T)^{-1} \] (18)

Each \( s_i \) is now detected as its permissible value at the minimum distance from the corresponding \( x_i \). This detection process is an arrangement of exact equalization followed by the application of suitable decision threshold to the equalized signal and is the equivalent but not the same as the linear transversal equalizer for the continuous system. This is referred to as Block Linear Equalizer (BLE). The BLE can be implemented through an iterative process using a simple piece of equipment [8].

V. COMPARISON OF BLOCK LINEAR EQUALIZER WITH LINEAR TRANSVERSAL EQUALIZER

It is well known that when one or more zeros of the z-transform of the sampled impulse response of the channel lie on the unit circle in the z-plane a linear transversal equalizer with finite number of tap gains, cannot equalize the channel correctly [1-3]. However, correct operation in every case is achieved through the transmission of signal in blocks. This demonstrates the one important advantage of the Block Linear Equalizer over the Linear Transversal Equalizer. The reason for this is that in the case of BLE the channel is equalized by a network rather than a linear transversal filter.

Although any channel with a finite impulse response can be equalized by the appropriate equalizer, for certain values of the channel sampled impulse response, particular sequences of transmitted element values result in no signal at the output of the receiver filter [1]. No amount of linear or non-linear equalization with continuous signal can give correct operation for the prolonged transmission of such sequences and unique detection of such a signal cannot normally be achieved in practice. In the case of BLE, it can be seen that the received signal vector \( SY \) in Eqn. (15) in the absence of noise, must be non-zero for all possible values of S. Thus in this case there is no complete loss of received signal whatever may be the sequence of the transmitted element values (assuming of course that the sampled impulse response of the channel is non-zero).

Since the LTE does not make use of the prior knowledge of the g zero level elements, its tolerance to noise is considerably reduced. Thus in order to make a more realistic comparison the performances of the LTE must be determined by assuming a continuous stream of signal without any gaps, which has the same information rate as the BLE. This means that for a given information rate the element transmission rate in the LTE, is reduced from \( 1/T \) to \( 1/T' \) where

\[ T' = [(m+g)/m] T \] (19)

Thus the signal-elements in the arrangement using continuous transmission, are now transmitted with no gaps at intervals of \( T' \) seconds and the received signal is sampled at the time instants \( t = iT' \) for all integer i.

The transmitter and receiver filters assumed for the BLE now have an unnecessary wide band width at the new sampling rate of \( 1/T' \) and introduce some inter-symbol interference. The transmitter and receiver filter characteristic must, therefore, be modified to \( h'(t) \) which is the equivalent value at the new sampling rate. It may be noted that while comparing the noise performance of the BLE and the LTE the transmission path in Figure 1, is to be the same in the two cases. Thus since the sampling rates are different in the two systems and since each system uses the transmitter and receiver filter appropriate to its sampling rate, the sampled impulse response of the base band channel corresponding to any given transmission path is...
different in the two systems. The impulse response \( y'(t) \) of the corresponding baseband channel for the LTE is determined by convolving \( h'(t) \) with the impulse response of the transmission path. The sampled impulse response of the baseband channel is then determined by sampling \( y'(t) \) at intervals of \( T' \) seconds, the sampling instants being phased so that one of the these coincides with the positive peak of \( y'(t) \). The phase of sampling instants selected is such that it is most likely to maximize the tolerance to additive white Gaussian noise [1-4].

The Bit Error Rate for the Block Linear equalizer is evaluated for channels A and B having the following sampled impulse responses

Channel A : [0.235 0.667 1.0 0.667 0.235]
Channel B : [-0.235 0.667 1.0 0.667 -0.235]

and for eight signal-elements in a block. Channel A is selected to give a large amplitude distortion while Channel B is selected to provide phase distortion. The performance of the LTE for the equivalent channel is also obtained. The results for the two systems are shown in Figures 2 and 3.

It can be seen from the Figure 2. Bit figures that the performance of the LTE in the presence of additive Gaussian noise and severe signal distortion (Channel A) is slightly better than that of the BLE. In case of pure phase distortion (Channel B) the noise performance of the two systems is almost same. However, it may be pointed out that the performance of LTE is here calculated neglecting the error extension effects. Furthermore, since \( g/m \) is large here than would be used in a practical system, the results indicate that in actual practice the noise performance of the Block Data Transmission System should be similar or even better than that of the Continuous System.

VI. CONCLUSIONS

Block Data Transmission System considered here has some useful advantages over the Continuous Transmission System. Firstly, exact equalization of the channel is, in every case, achieved. Secondly, complete loss of signal cannot result from an unfortunate combination of signal-element values and channel impulse response. Thirdly there are no error extension effects from one group of elements to the next, regardless of the detection process used. Finally, detection process achieving a high tolerance to additive Gaussian noise can be implemented quite simply. The disadvantage of BDTS is that for a given information rate, the bandwidth required is wider than that required for the Continuous System. This reduces the tolerance to noise of the BLE and partly offsets the basic advantages gained by the arrangement. However, when the number of elements in a group is relatively large compared with the number of elements set to zero between adjacent groups, a useful advantage in tolerance to noise should be gained by the Block Linear Detector over the Linear Transversal Equalizer, where the latter is used with a continuous (uninterrupted) signal having the same information rate.

REFERENCES


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